

REMARKS

In view of the following discussion, the Applicants submit that none of the claims now pending in the application are unpatentable under the provisions of 35 U.S.C. § 103. The Applicants herein amend claims 1 and 16. Support for the amendments may be found in the Applicants' specification on at least paragraphs [16] and [18]. Thus, the Applicants believe that all of these claims are now in allowable form.

I. REJECTION OF CLAIMS 1-4, 7-19 AND 23-31 UNDER 35 U.S.C. § 103

A. Claims 1-4, 7-11, 16-19, and 23-27

The Examiner rejected claims 1-4, 7-11, 16-19, and 23-27 as being unpatentable over Summers, et al. (U.S. Patent No. 6,961,416, issued on November 1, 2005, hereinafter referred to as "Summers") in view of Rodman, et al. (U.S. Patent Publication No. 2002/0103864, published on August 1, 2002, hereinafter referred to as "Rodman") and in further view of Aravamudan, et al. (U.S. Patent No. 6,584,076, issued on June 24, 2003, hereinafter referred to as "Aravamudan"). The Applicants note that the Examiner included claims 20-22 in the rejection. However, the Applicants note that claims 20-22 were previously canceled without prejudice. Thus, the Applicants respectfully traverse the rejection.

Summers teaches an internet-enabled conferencing system and method accommodating PSTN and IP traffic. A caller may call into a conference call by dialing a number connecting them to a Voice node or VoIP node within a chassis on a TDM bus. (See Summers, col. 11, ll. 26-65).

Rodman teaches a system and method for coordinating a conference using a dedicated server. The system and method initiates a data conference between a plurality of conference endpoints linked in communication by a private or public computer network. (See Rodman, Abstract).

Aravamudan teaches a telecommunications conferencing method and apparatus. The method and apparatus use a plurality of device servers including a packet circuit gateway. In response to a request for a conference call, the

packet network determines the parties to be on the conference call and selects a conference bridge that results in the lowest cost for the conference call. (See Aravamudan, Abstract).

The Examiner's attention is directed to the fact that Summers, Rodman and Aravamudan, alone or in any permissible combination, fail to teach or to suggest a method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection, setting up a connection between said VCS and said first VoIP station and receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up, as positively claimed by the Applicants' independent claims 1 and 16, respectively. Specifically, Applicants' independent claims 1 and 16 recite:

1. A method for establishing a Voice over Internet Protocol (VoIP) conference call by joining a first VoIP station in a communication between a plurality of communication stations, wherein at least one of the plurality of communication stations is a second VoIP station in a private network and said first VoIP station is in the private network, the method comprising:

receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection;

setting up a connection between said VCS and said first VoIP station;

receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up, wherein said one of said plurality of existing conversations is between the second VoIP

station in the private network and a phone in a public network, wherein said VCS is external to said first VoIP station and said plurality of communication stations;

establishing a Real-Time Transport Protocol (RTP) voice path with the first VoIP station and said VCS; and

managing data packet transmission between the first VoIP station and one of the plurality of communication stations via said VCS.
(Emphasis added).

16. A device for establishing a Voice over Internet Protocol (VoIP) conference call by joining a first VoIP station in a communication between a plurality of communication stations, wherein at least one of the plurality of communication stations is a second VoIP station in a private network and said first VoIP station is in the private network, the device comprising:

a receiver in a Voice Conference Server (VCS) for receiving an indication from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS after setting up a connection between said VCS and the first VoIP station, wherein each one of said plurality of existing conversations is on a different connection, wherein said indication comprises a code number entered by said first VoIP station corresponding to the second VoIP station identifying said one of said plurality of existing conversations on said VCS, wherein said one of said plurality of existing conversations is between the second VoIP station in the private network and a phone in a public network, wherein said VCS is external to said first VoIP station and said plurality of communication stations;

an apparatus in said VCS for setting up a Real-Time Transport Protocol (RTP) voice path with the first VoIP station in response to the received signal for joining said call; and,

an RTP mixer in said VCS for managing at least two VoIP stations and sending the mixed data packets to at least one VoIP station.
(Emphasis added).

In one embodiment, the Applicants teach a method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection, setting up a connection between said VCS and said first VoIP station and receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by

said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up. As a result, the Applicants' disclosure provides conferencing capability in private VoIP networks while containing costs for the VoIP phones because of the VCS. (See e.g., Applicants' specification, p. 6, para. [15]). Moreover, the VCS may provide conferencing capabilities without the need to pre-establish a conference call. (See *Id.* para. [16]).

The alleged combination (as taught by Summers) fails to render obvious the Applicants' claims because the alleged combination fails to teach or suggest a method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection, setting up a connection between said VCS and said first VoIP station and receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up. The Applicants have amended the claims to specify that the first VoIP station first establishes a connection with the VCS and then identification of a desired existing conversation is provided to the VCS. As a result, no conference call bridge needs to be pre-established.

In stark contrast, all of the references cited by the Examiner only teach traditional methods of conference calling. That is, each reference requires a conference bridge to be pre-established and the code provided allows the invited participants to connect to the conference server. Notably, once the invited participant has connected to the conference server, the invited participant is unable to then select a conversation. This significant difference is clear upon examination of the amended claim language recited in the Applicants' independent claims 1 and 16.

Summers clearly teaches that a conference is scheduled and that once the scheduled start time arrives that callers join the conference using a dial in or dial out procedure. (See Summers, col. 11, ll. 26-31). For example, the caller dials a conference telephone number or IP address previously provided to the caller. (See *Id.* at ll. 36-40, emphasis added).

In stark contrast, the Applicants' method does not require a user to dial a conference call number. Rather, the user in the Applicants' method simply sends an indication to the VCS that they desire to join an existing conversation.

In addition, Summers teaches away from the Applicants' method because Summers teaches that each caller must dial into the same conference call telephone number. Moreover, Summers fails to teach or suggest that once the callers are connected to the conference server that the callers can then select an existing conversation.

In stark contrast, the Applicants' method teaches setting up a connection between said VCS and said first VoIP station and receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up. In other words, multiple communication stations may carry a conversation via the VCS to allow a user to join any one of the existing conversations after the caller connects to the VCS. To do so, the user simply needs to identify which connection the user wishes to join via the VCS.

Moreover, Summers fails to teach or suggest receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up. At best, the conference call numbers taught by Summers identify a pre-established conference call and not a second VoIP station. In addition, the Examiner concedes that Summers fails to teach or suggest this limitation. (See Final Office Action, p 3, ll. 14-20).

In addition, Rodman fails to bridge the substantial gap left by Summers because Rodman also fails to teach or suggest receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection, setting up a connection between said VCS and said first VoIP station and receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up. As previously noted, Rodman teaches away from the Applicants' method because Rodman teaches that the initial audio conference is established by dialing a telephone number and entering a code assigned to the conference bridge. (See Rodman, para. [0038]). The codes referred to by the Examiner are used with a data conference that may be initiated only for those participants already in the existing audio conference. (See Rodman, para. [0038]-[0039]).

Moreover, the data conference codes used in Rodman are for allowing an invited participant to join a newly generated data conference. (See Rodman, para. [0047]). Notably, the caller is unable to identify an existing conversation after they connect to the conference bridge.

Finally, Aravamudan also fails to bridge the substantial gap left by Summers and Rodman because Aravamudan also fails to teach or suggest receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection, setting up a connection between said VCS and said first VoIP station and receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second

VoIP station after said connection between said VCS and said first VoIP station is set up. Aravamudan only teaches dynamically changing conference bridges. (See Aravamudan, Abstract). Therefore, Summers, Rodman and Aravamudan, alone or in any permissible combination, fail to render obvious the Applicants' independent claims 1 and 16.

Moreover, dependent claims 2-4, 7-11, 17-19, and 23-27 depend, either directly or indirectly, from independent claims 1 and 16, respectively, and recite additional limitations. As such, and for the exact same reason set forth above, the Applicants submit that claims 2-4, 7-11, 17-19, and 23-27 are also patentable over Summers, Rodman and Aravamudan. As such, the Applicants respectfully request the rejection be withdrawn.

B. Claims 12-15 and 28-31

The Examiner rejected claims 12-15 and 28-31 as being unpatentable over Summers in view of Rodman and Aravamudan and in further view of Canon, et al. (U.S. Patent No. 6,269,159, issued on July 31, 2001, hereinafter referred to as "Cannon"). The Applicants respectfully traverse the rejection.

The teachings of Summers, Rodman and Aravamudan have been discussed above. Cannon teaches conferencing with a calling party. The method and apparatus provides three way conferencing which allows a third party caller to call into an existing telephone call at a single line of a called party's telephone. (See Cannon, Abstract.)

The Examiner's attention is directed to the fact that the alleged combination (as taught by Summers, Rodman, Aravamudan, and Cannon) fails to disclose the novel a method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection, setting up a connection between said VCS and said first VoIP station and receiving an

identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up, as positively claimed by the Applicants' independent claims 1 and 16.
(See *supra*).

The Applicants' method teaches a method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection, setting up a connection between said VCS and said first VoIP station and receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up. In contrast, as discussed above, the combination of Summers, Rodman and Aravamudan simply does not teach or suggest the novel method or apparatus for establishing a VoIP conference call comprising receiving an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection, setting up a connection between said VCS and said first VoIP station and receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up.

Moreover, Cannon does not bridge the substantial gap left by Summers, Rodman and Aravamudan because Cannon also fails to teach or suggest a method or apparatus for establishing a VoIP conference call comprising receiving

an indication at a Voice Conference Server (VCS) from the first VoIP station in the private network for joining one of a plurality of existing conversations on said VCS between the plurality of communication stations, wherein said plurality of existing conversations are VoIP calls and each one of said plurality of existing conversations is on a different connection, setting up a connection between said VCS and said first VoIP station and receiving an identification of said one of said plurality of existing conversations on said VCS via a code number entered by said first VoIP station corresponding to the second VoIP station after said connection between said VCS and said first VoIP station is set up. Cannon only teaches a method and apparatus for conferencing with a calling party. (See Cannon, Abstract). Thus, for all of the above reasons, the Applicants respectfully contend that the combination of Summers, Rodman, Aravamudan and Cannon fails to render obvious the Applicants' independent claims 1 and 16.

Furthermore, dependent claims 12-15 and 28-31 depend, either directly or indirectly, from claims 1 and 16, respectively, and recite additional limitations. As such, and for the exact same reason set forth above, the Applicants submit that claims 12-15 and 28-31 are also patentable and not made obvious by the teachings of Summers, Rodman, Aravamudan and Cannon. As such, the Applicants respectfully request the rejection be withdrawn.

CONCLUSION

Thus, the Applicants submit that all of these claims now fully satisfy the requirements of 35 U.S.C. § 103. Consequently, the Applicants believe that all these claims are presently in condition for allowance. Accordingly, both reconsideration of this application and its swift passage to issue are earnestly solicited.

If, however, the Examiner believes that there are any unresolved issues in any of the claims now pending in the application, it is requested that the Examiner telephone Mr. Kin-Wah Tong, Esq. at (732) 842-8110 so that appropriate arrangements can be made for resolving such issues as expeditiously as possible.

Respectfully Submitted,

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